ZoIPER

USER GUIDE

Edition Version 2.0 for Windows®

Date of issue: July 2007

0. Notice

Liability

All rights reserved.

- This *User Guide* is intended to demonstrate typical uses and features of the ZOIPER Biz SIP and IAX/IAX2 soft phone: Version 2.0. It is up to the user of this manual to decide whether the information mentioned herein is applicable to the particular IP-based network (s)he wants to run this soft phone application on.

- The ZOIPER team and persons involved in the composition of this *User Guide* will in no case be held responsible for any incidental, indirect or otherwise consequential damage or loss that may result from using ZOIPER Biz Version 2.0.

Conventions used in this document

The following typographical conventions are used in this document:

- Hyperlinks to sections inside this document, email and the Internet are underlined and blue.
- The names of windows are marked in ***Bookman Old Style, bold and italics***.
- The names of files, directories and syntax of commands are written in *italics.*
- Parameters of commands are written as follows: <*italics*>
- Buttons start with a capital letter.
- Sequential clicking on buttons and names of (sub)directories are separated by an arrow pointing to the right: **>**
- Sequential steps in a process are preceded with numbers: 1,2,3...

CONTENTS

0. NOTICE **2**

Liability 2
Conventions used in this document 2

1. ZOIPER EDITIONS **5**

Introduction 5
ZOIPER Free 6
Personalised ZOIPER 7

2. GETTING STARTED WITH ZOIPER **8**

System requirements 8
A glance at ZOIPER 8
Download 9
Installing ZOIPER 9
Uninstalling ZOIPER 9
Starting ZOIPER for the first time 10
Creating users in Asterisk® 10

3. CALLING WITH ZOIPER **12**

Selecting an account 12
Outgoing calls 12
Selecting a contact person *12*
Selecting a phone line *14*
Calling a contact person *14*
Ending a phone call 14
Incoming calls 14
Hold 15
Transferring calls 16
Transfer *16*
Attended transfer *16*
Missed calls 17
Forwarding calls/Auto Answer 17
Conference & Voice mail 17

4. ADDRESS BOOK **18**

Accessing the address book 18
Contact information 18
Adding and deleting entries 18
New entry *18*
Deleting entries *19*
Editing entries 19
Sorting entries 19

5. VOLUME CONTROL **21**

6. LOG **22**

7. HISTORY **23**

Accessing the History 23
Call information 23

8. OPTIONS **25**

Accessing the options 25

Options menu *25*
SIP ACCOUNTS 26
Add a new SIP account *26*
SIP Advanced account options *28*
IAX ACCOUNTS 29
Add a new IAX account *29*
IAX Advanced account options *31*
PROTOCOL OPTIONS 32
SIP options *33*
IAX options *33*
RTP options *33*
- *Advanced RTP options* *34*
STUN Options *34*
- *Advanced STUN options* *35*
AUDIO OPTIONS 35
General *35*
Audio devices *36*
- *Audio devices advanced options* *37*
Audio codecs *37*
GENERAL OPTIONS 38
General *38*
Call events *39*
LANGUAGES 40
UPGRADE TO ZOIPER BIZ 40
ABOUT 40

1. ZOIPER Editions

Introduction

This document will guide you step by step to your actual aim: reducing the costs of your phone calls by using the VoIP (Voice over Internet Protocol) technology of *Asterisk®* and the latest version of ZOIPER soft phone.

In the rest of the first chapter of this *User Guide* you can find more details about the ZOIPER Biz Edition. If you are interested in having your "own" phone but don't have the resources to have it developed, there is the option of ordering a re-branded OEM ZOIPER.

Do not hesitate to send a mail to zoiper@asteriskguru.com for more information and offers. Our team of experienced professionals is most willing to answer all your questions.

At the same time, our developers are looking forward to implement your specific desires in future versions of ZOIPER. You can contact them at support@asteriskguru.com

Together, we make calling more comfortable.

ZOIPER Free

ZOIPER Free Edition V.2.0 has many features to provide you with a very comfortable calling experience:

- SIP + IAX / IAX 2 protocols
- Available codecs – GSM, ulaw, alaw, speex, ilbc
- STUN support
- STUN server per account
- Two accounts
- DTMF tones sending
- Echo cancellation
- Codec settings per account
- Account password encryption
- Call history
- Hold function
- Quick dial panel
- Optional Automatic pop-up window for incoming call
- Call logs
- Minimize on tray
- Minimize on start up
- Always on top
- Adaptive Jitter Buffer
- Support for multiple audio devices
- Address book
- Quickdial pad
- Automatic user registration
- Call transfer
- Voice mail message information
- Portable ZoIPer with portable devices (like USB, flashcards, etc.)
- Multilanguage support

Personalised ZOIPER

- Do you want your employees to use state-of-the-art VoIP technology with your own company's logo and colors clearly visible on it?
- Do you want to surprise your clients with your “own” personal telephone?

You can opt for a personalised verion of ZOIPER Biz Edition. The basic rebranding includes:

- changing the name “ZOIPER” to your choice.
- changing the color scheme to your choice.
- changing the logo to the logo of your choice.

You may also go for the complete customization of ZOIPER Biz Edition. Apart from change of colors and logo, this includes complete change of interface (adding/removing buttons/features) and functionality!
Contact zoiper@asteriskguru.com for more information.

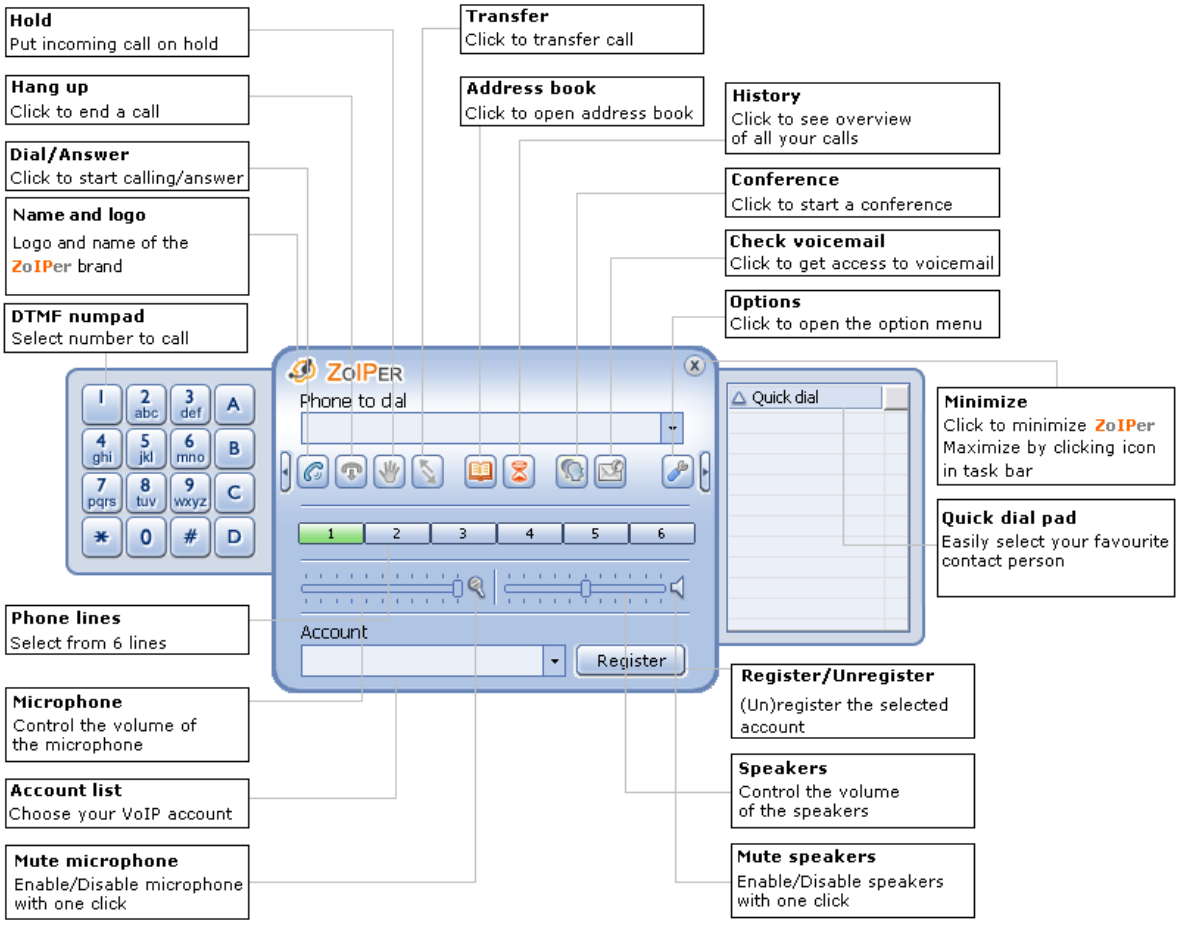
2. Getting started with ZOIPER

System requirements

In order to use ZOIPER properly, you need at least following system configuration:

- Processor: minimum *Pentium* II 300
- Memory: minimum 256 MB RAM
- Operating systems: *Windows*® 2000, XP and later
- Sound card: 16 bit sound card (SoundBlaster or equivalent)
- Internet connection: wired or wireless broadband

A glance at ZOIPER



You can see the function of the dial buttons when you hold your mouse over the button.

Download

You can read more information about ZOIPER Free Edition and download it here.

Installing ZOIPER

To install ZOIPER, follow the instructions below:

1. Click on the icon of the downloaded *Zoiper2.0 Free Installer.exe* file
2. The ***InstallShield Wizard*** now opens. Click Next, read our EULA and if you agree with the terms and conditions, click I agree.
3. Choose the destination folder in which to install ZOIPER (default is C:\Program Files\Attractel\Zoiper Free) by clicking on the Next button.
4. Choose who will be making VoIP calls with ZOIPER (All users or Current user), and click Next>Next >Finish.

Uninstalling ZOIPER

To remove ZOIPER from your computer, choose one of the following:

- From the Windows Start Menu>Programs>Attractel>Zoiper Free>Uninstall Zoiper Free.
- Via the Control Panel>Add/Remove Programs. Find ZOIPER from the list of applications, click on the Remove button.
- Follow these steps:

    1. Click on the icon *Zoiper 2.0 Biz installer.exe.*
    2. The ***InstallShield Wizard*** now opens. Click Next.
    3. Tick the third option *Remove* and click on Next.

    4. Click on the Remove button.
    5. Click on Finish.

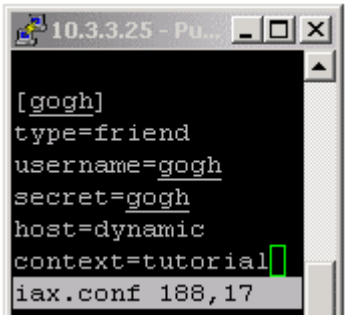
Starting ZOIPER for the first time

- Open ZOIPER click Start > Programs > Attractel > Zoiper Free. You can now start using the ZOIPER softphone.

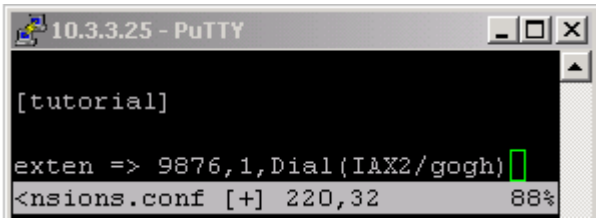
Creating users in Asterisk®

1. The first thing to do is create users in the iax.conf file. You have to provide values for the following fields:

Field	***Description***	***Example***
type	You can choose from 3 different *types:* ***friend****:* make and receive calls ***user****:* can only make calls ***peer****:* can only be called	"friend"
username	Used for authentification.	"gogh"
secret	The password of this user.	"gogh"
host	Dynamic or static.	"dynamic"
context	The environment in which the user can make and receive calls.	"tutorial"



2. In the extensions.conf file, you link the created users to an *extension* and a *context*. In the example below, the user "Gogh" can call and be called in the context "tutorial" with the Dial application. His extenion is "9876".



3. Reload Asterisk from CLI.

For more information about how to configure the Asterisk® configuration files, please read our tutorial about Configuring IP Phones for Asterisk.

3. Calling with ZOIPER

Selecting an account

- You can select the account you want to call with, by choosing the desired account from the dropdown menu.

- To register an account:
  - Select a user from the list
  - Click on Register

- To unregister an account:
  - Select a user from the list.
  - Click on Unregister

- All your registered accounts can be dialed to.

Outgoing calls

Selecting a contact person

There are different ways to select the user you want to call:

- place the cursor in the *Phone to dial* field and type the extension on your keyboard.

- open the Address book. Double click on the person you want to call.

- open the ***DTMF numpad*** by clicking on the small arrow pointing to the left.

You can now compose the dial extension with your mouse by clicking on the DTMF buttons.

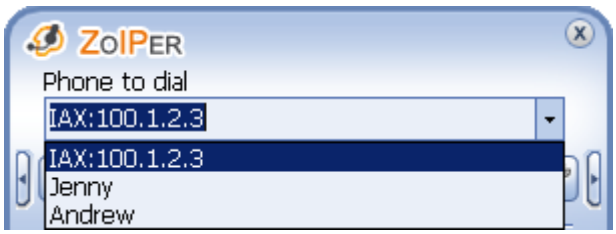
- enter a dial string in the field *Phone to dial,* as in the example below. The dial string is composed as follows:
  - SIP

  sip: <user>:<password>@<servername>/<extension> @<realm>
  - IAX

  iax: <user>:<password>@<servername>/<extension> @<context>

- Place the cursor in the *Phone to dial* field and type in the name and the IP address/servername of the callee and click on the Dial/Answer button. You do not need to have a selected account for direct calling.

- click on the arrow of the *Phone to dial* drop-down list. Here you can find the persons you have called in chronological descending order.



- click on the small arrow pointing to the right The ***Quick dial pad*** opens.
  If you double click on a contact person, you will start calling him.

Have a look at this paragraph to find out how to add contact persons to the ***Quick dial pad***.

Selecting a phone line

- You can make use of 6 phone lines for both incoming and outgoing conversations.
- You can easily switch from lines by left clicking on the Line button you want to use for the phone call.
- The selected line by default is Line 1 and its colour is green. When you select any other idle line (e.g any blue line) it becomes selected and green.
- A line in a middle of an established conversation is coloured in yellow. If you select it, its colour changes to green.
- A line, receiving an incoming call becomes pink in colour. Note that an incoming call is also notified with an ***Info Pad***, and an optional pop up window.

Calling a contact person

- If you have selected the person you want to call and which line to use, you can call your contact person by clicking on the Dial/Answer button. The Hang up button will turn red now.

Ending a phone call

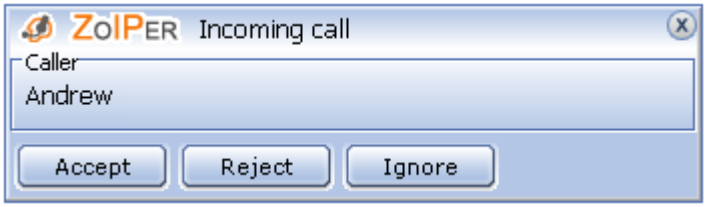
To end a conversation, simply click on the Hang Up button

Incoming calls

When you have an incoming call, ZOIPER notifies you in the following ways:

- You hear a ringtone in your headphones/speakers.
- An *Info pad* appears on top of ZOIPER. To learn more about the *Info pad*, click here.
- The first idle line will become pink.

- If you want to, you can also choose for a *Pop-up menu* with the *callerID* of the caller, as shown below. With this pop-up window, you have 3 options.



  - You can choose to Accept the call. You can also accept a call by clicking on the Dial/Answer button.
  - You can choose to Reject the call. You can also reject a call by clicking on the Hangup button.
  - Clicking on Ignore will not end the call. It will just ignore the pop-up screen. The *Info pad* will remain on top of ZOIPER and the ringtone will go on.

Check the Call events options to learn how to receive a pop-up window for incoming calls.

Hold

- When an incoming or outgoing call becomes accepted, the Hold button will become enabled.

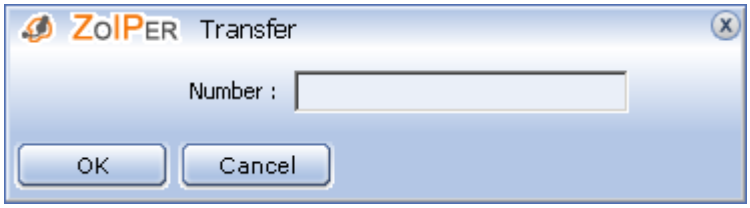
- You can configure Asterisk so that the caller who is put on hold listens to music while waiting on the phone.

Transferring calls

Transfer

1. When an incoming or outgoing call becomes accepted, the Transfer button will become enabled. To transfer a call (no matter incoming or outgoing), click on the Transfer button.

2. The *Transfer* pop-up window will appear.



Enter the recipient number of the call to be transfered in the *Recipient* field. Already entered recepients cn be seen by clicking on the drop down button.

Attended transfer

To make an Attended transfer with ZOIPER:

1. Change the Asterisk configuration file features.conf to the desired DTMF sequence that will be used for Attended transfer.
2. Find in the configuration file the lines shown in the example below, to uncomment the lines and set them to the desired values.

- Blind transfer will be set to be executed after you press '#' in a call.
- Attended transfer will be executed on pressing '*'.
- To use the '#' sequence for attended transfer a different sequence should be set for blind transfer sequence.

  - ***Example:***

features.conf

[featuremap]

blindxfer => # ; Blind transfer

*atxfer => * ; Attended transfer*

To transfer calls using DTMF sequences the 't' and 'T' options should be set into the Dial() application parameters in extensions.conf. This allows the called and the caller party to transfer the call by sending the DTMF sequence defined in features.conf.

- ***Example:***

extensions.conf

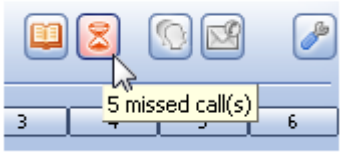
----

exten => 555010203,1,Dial(IAX2/some-user,t,)

To make an attended transfer enter the sequence defined in the atxfer field. To finish successfully the attended transfer the transferring side should hang up.

Missed calls

- If you have a missed call, the History button will be flashing red. If you move the mouse over the button, the number of missed calls will be shown.

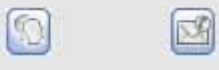


- To see more details about the missed call in the History, click on the History button.

Forwarding calls/Auto Answer

The Call forwarding/Auto answer feature is available with ZOIPER Biz only.

Conference & Voice mail

Conference and Voice mail are options for ZOIPER Biz. These functions are not available with ZOIPER Free. Therefore, both buttons are disabled: 

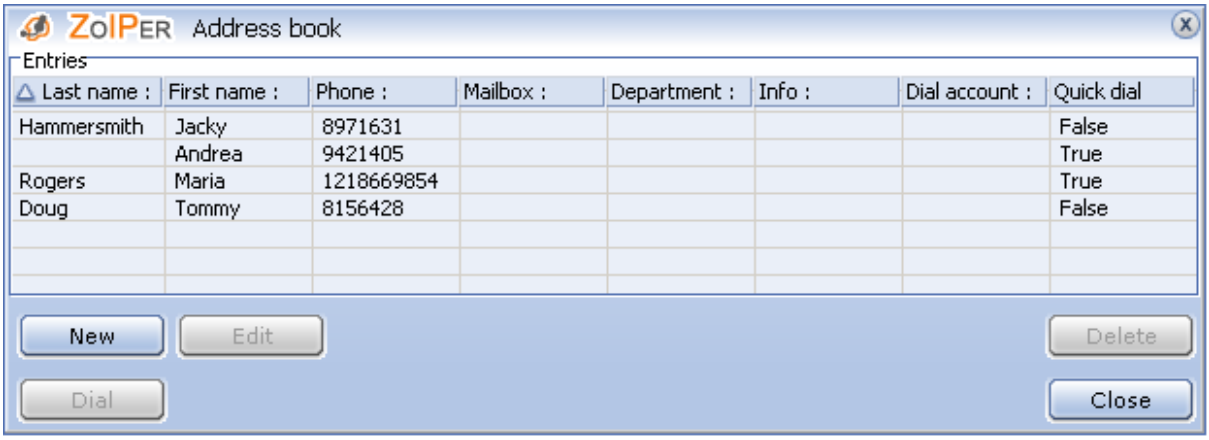
4. Address book

Accessing the address book

Open your ***Address book*** by clicking on the Address book button.

Contact information

- You can enter useful information about each contact person in 7 different fields: *Last name, First name, Phone, Mailbox, Department, Info* and *Dial account.*



- Provide info for at least *First name* and *Phone.* The other 5 fields can be left empty if you want to.

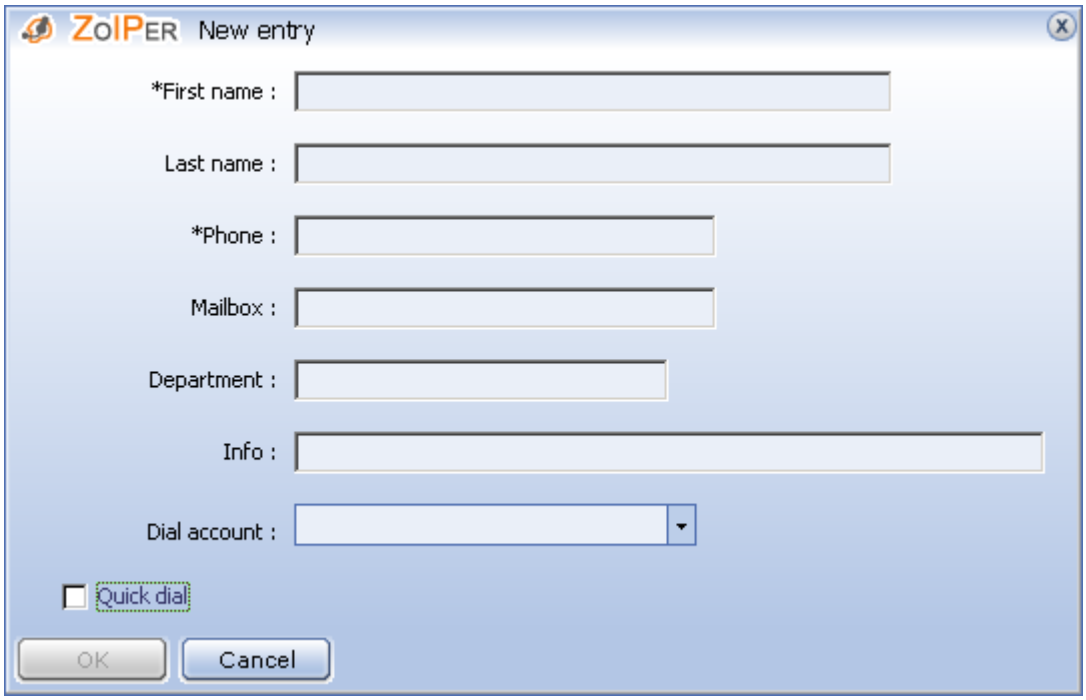
To save time, it is best to add the persons you call most often to the ***Quick dial pad***. You can do this by ticking the *Quick dial* check box when you add a new entry to your ***Address book***.

Adding and deleting entries

New entry

1. Click on the **New** button.

2. Enter information in the ***New entry*** window for at least following fields: **First name* and **Phone extension.* The other fields are added for your convenience and are non-mandatory.



3. Save the entry by clicking on the enabled OK button.

Deleting entries

1. Click on the entry you want to delete from your ***Address book***.
2. Click on the Delete button.

Editing entries

1. Click on the entry you want to change.
2. Click on the Edit button.
3. Edit the information for this entry.
4. Store your changes by clicking on OK.

Sorting entries

The contact persons can be sorted on the 8 different fields: *Last name, First name, Phone, Mailbox, Department, Info, Dial account* and *Quick dial.*

1. Click on a field and an arrow will appear like in the example below: *Last name*.
2. Click again on the arrow to sort the list in ascending or descending order.

5. Volume control

- You can easily control the incoming and outgoing volume of ZOIPER. Adjust the volume by moving the sliders to the left (less sound) or the right (more volume).

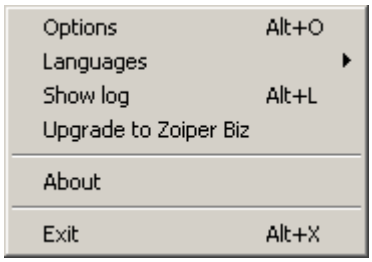
- If you want to disable/enable the incoming/outgoing sound, click on the icon of the Speaker, respectively the Microphone. The icons will turn red when the sound is mute.

- Also note the Audio level indicators within the sliders. They indicate the outgoing and incoming sound respectively.

6. Log

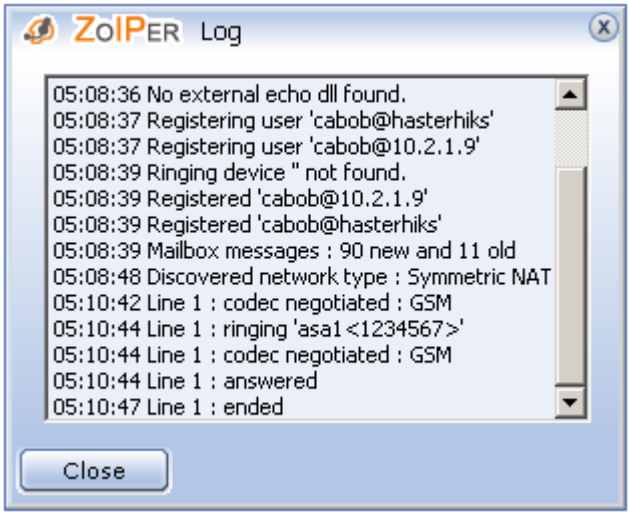
To open the ***Log***, you must either:

- right-click on ZOIPER with your mouse and choose *Show Log* from the popup menu.



- press Alt+L on your keyboard.

The ***Log*** window contains useful detailed information about the sequence of events, and the time they took place. This information refers to internal ZOIPER processes as well, which makes it very useful for more advanced users.

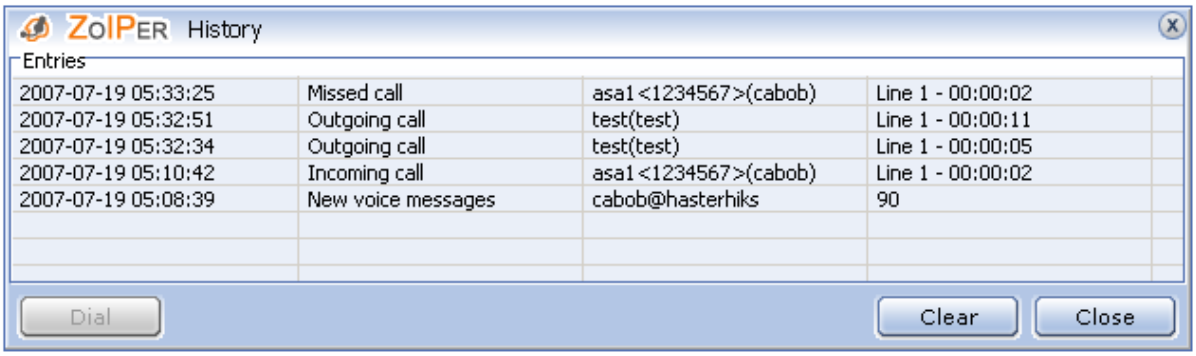

ZoIPER Log
05:08:36 No external echo dll found.
05:08:37 Registering user 'cabob@hasterhiks'
05:08:37 Registering user 'cabob@10.2.1.9'
05:08:39 Ringing device '' not found.
05:08:39 Registered 'cabob@10.2.1.9'
05:08:39 Registered 'cabob@hasterhiks'
05:08:39 Mailbox messages : 90 new and 11 old
05:08:48 Discovered network type : Symmetric NAT
05:10:42 Line 1 : codec negotiated : GSM
05:10:44 Line 1 : ringing 'asa1<1234567>'
05:10:44 Line 1 : codec negotiated : GSM
05:10:44 Line 1 : answered
05:10:47 Line 1 : ended
Close

7. History

Accessing the History

You can open the ***History*** by clicking on the History button.

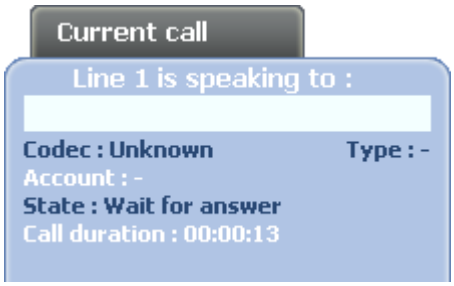
- When in the ***History*** window you click on an entry of a *Missed call*, you can call your contact person by clicking on the enabled Dial button. You can also scroll up and down to see the chronological sequence of events.
- You can also choose to Clear the content or Close the ***History*** window.


ZoIPER History
Entries
2007-07-19 05:33:25 Missed call asa1<1234567>(cabob) Line 1 - 00:00:02
2007-07-19 05:32:51 Outgoing call test(test) Line 1 - 00:00:11
2007-07-19 05:32:34 Outgoing call test(test) Line 1 - 00:00:05
2007-07-19 05:10:42 Incoming call asa1<1234567>(cabob) Line 1 - 00:00:02
2007-07-19 05:08:39 New voice messages cabob@hasterhiks 90
Dial
Clear
Close

- For each phone call the ***History*** contains the following information:
  - Date and time of calling.
  - Status of the call. This is one of the following:
    - missed
    - unanswered
    - new voice mail messages
    - incoming
    - outgoing
  - Name/Number of called/calling person.
  - Phone line used for the call.

Call information

Every time you dial or receive a call, an *Info pad* pops on top of the ZOIPER window.



The *Info pad* is showing information about:

- The callerID of the caller party
- The codec the caller is using
- The type of protocol used for the call
- The account that receives the call
- The state the current line is in. This could be one of the following:
    - Up – when you or the others side picks up the call
    - Down – when you or the other side hangs up the call
    - Ringing – when you or the other side is still ringing
    - Wait for Answer – when you have dialled and wait for the other side to respond
    - Active – the line is in active state when early media is detected
    - Dialling – when you are in a state of dialling before being connected
    - Resolving Port – the line is in a state of resolving 1the port for STUN
- Call duration – the time elapsed since the Info pad appears

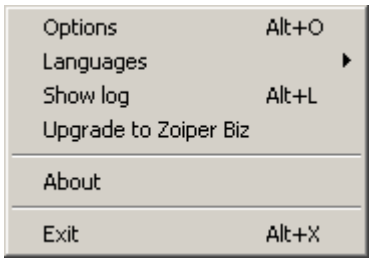
8. Options

Accessing the options

Options menu

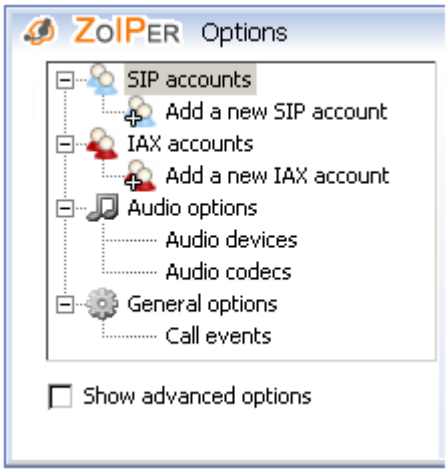
You can access the Options Menu of ZOIPER in three ways:

- right click on the ZOIPER icon in your system tray or on the phone itself and the following pop-up menu appears



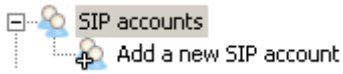
- press Alt+O on your keyboard
- click the Options button

In all cases the following screen will pop up:



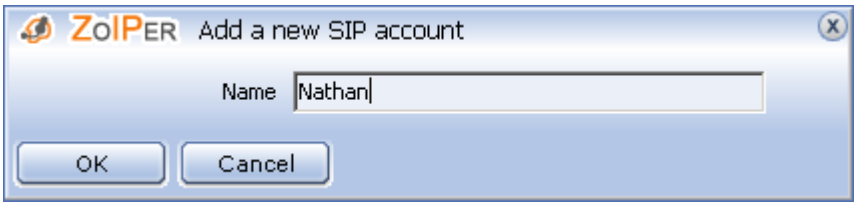
Click on the option in the *Option Tree* you want to change the settings for.

SIP accounts



Add a new SIP account

Enter an account name and press the OK button.



Each new account is added under the SIP accounts options in the *Options Tree*.To delete an existing SIP account, click on the account name in *Options Tree* and press "Del" on your keyboard. A confirmation popup will appear:

After entering an account name, press the OK button. Now you can access the SIP account options.

The ***SIP account options*** look as follows:

Domain

Enter the host name, specified by your VoIP service provider.

Username

Enter the username given to you for registration/authorization.

Password

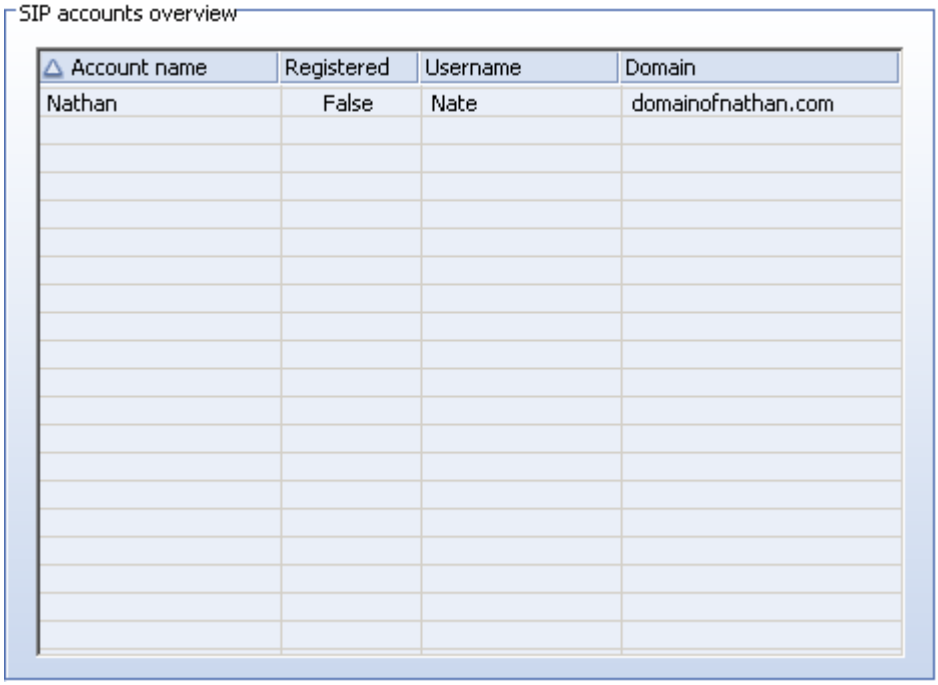
Enter the password given to you for registration/authorization.

Caller ID Name

Enter your Caller ID name. The callee side, if capable, will be seeing this Caller ID name whenever you call.

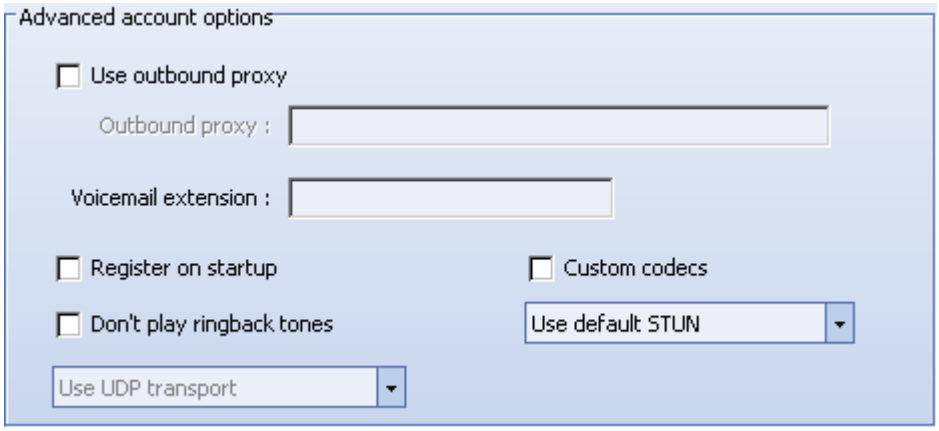
Do not forget to apply all changes by clicking on the Apply button.

Please mind that all added ***SIP accounts*** are listed in the ***SIP accounts overview*** table. As clearly visible from the picture below, it displays the *Account name*, *Username* and *Domain*, as well as whether the chosen account is *Registered* (True) or not (False). The SIP accounts overview is accessible by clicking on SIP accounts in the Options Tree. Note that all fields (except for Registered) could be sorted alphabetically or counter-alphabetically. The Registered accounts may be sorted by True or False.



SIP Advanced account options

You can enable/disable the Advanced account options by ticking/unticking the Advanced options checkbox at the bottom of the ***Options*** window.



Use outbound proxy

This option is for outgoing calls through a proxy server. Enter the host name or the IP address of the desired outbound proxy in the field below.

Voicemail extension

Enter the extension at which to check your voicemail messages.

Register on startup

This option is for automatically registering the current account each time ZOIPER starts up.

Don't play ringbacktones

Tick this checkbox to mute all ringbacktones.

Custom codecs

You can choose to save the customised selection and order of codecs for this account. The Audio codecs for the chosen account are handled quite like in the general Audio codecs.

- The dropdown menu at the bottom right reveals the following options:

Use default STUN

Choose the STUN server, set in the Protocol options. To set the default STUN server, go to the STUN options in the Advanced Protocol options.

Use Custom STUN

Choose a custom STUN server. The STUN options for the chosen account are handled quite like in the general STUN options.

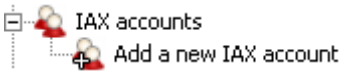
Don't use STUN

This option is for the case in which you do not need a STUN server for this account.

Do not forget to apply all changes by clicking on the Apply button. After applying the changes, the Audio codecs options and the STUN options for the current account appear underneath its name in the *Options Tree*.

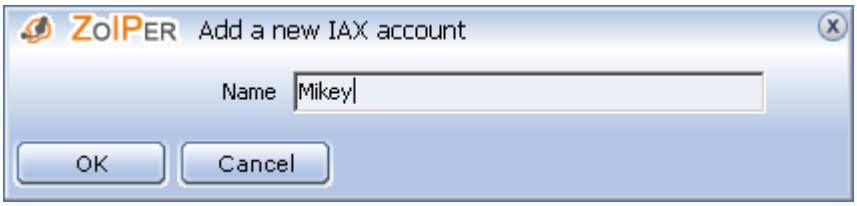
> The **TCP** and **TLS transport** options are features of ZOIPER Biz. These options are not available with ZOIPER Free. Therefore, only **UDP transport** is enabled.

IAX accounts

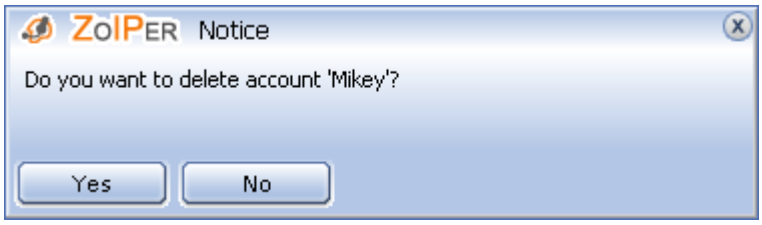


Add a new IAX account

Enter an account name and press the OK button.

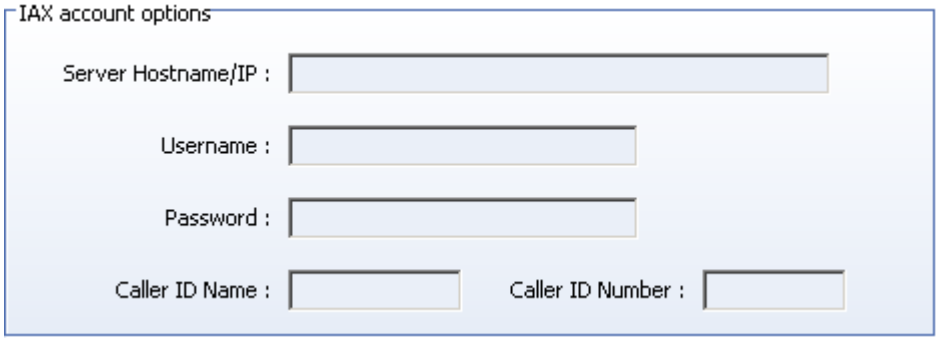


> Each new account is added under the IAX accounts options in the *Options Tree*.To delete an existing SIP account, click on the account name in *Options Tree* and press "Del" on your keyboard. A confirmation popup will appear:



After entering an account name, click the OK button. Now you can access the IAX account options.

The ***IAX account options*** look as follows:



Server Hostname/IP

Enter the IP address of your VoIP PBX or the IP address given to you by your VoIP Service Provider.

Username

Enter the username given to you for registration/authorization.

Password

Enter the password given to you for registration/authorization.

Caller ID Name

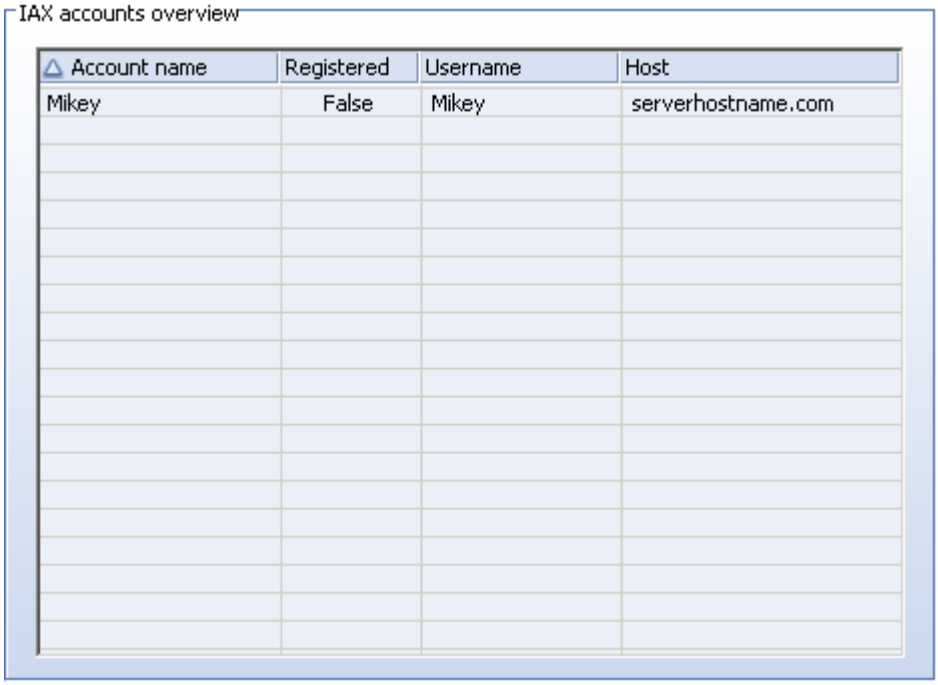
Enter your Caller ID name. The callee side, if capable, will be seeing this Caller ID name whenever you call.

Caller ID Number

Enter your Caller ID number. The callee side, if capable, will be seeing this Caller ID number whenever you call.

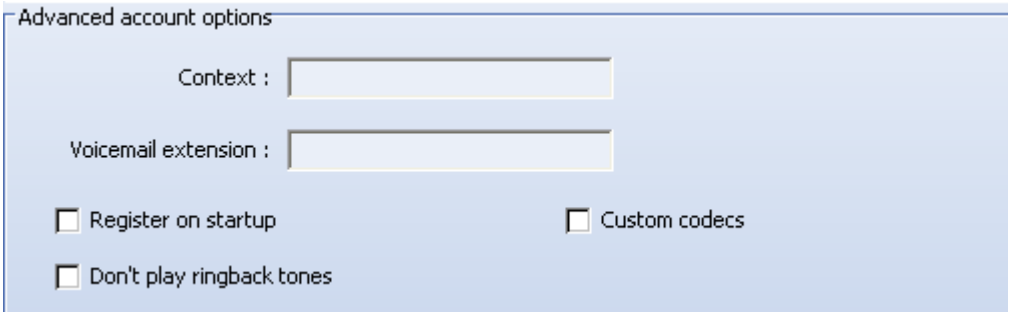
Do not forget to apply all changes by clicking on the Apply button.

Please mind that all added ***IAX accounts*** are listed in the ***IAX accounts overview*** table. As clearly visible from the picture below, it displays the *Account name*, *Username* and *Host*, as well as whether the chosen account is *Registered* (True) or not (False). The SIP accounts overview is accessible by clicking on SIP accounts in the Options Tree. Note that all fields (except for Registered) could be sorted alphabetically or counter-alphabetically. The Registered accounts may be sorted by True or False.



IAX Advanced account options

You can enable/disable the ***Advanced account options*** by ticking/unticking the Advanced options checkbox at the bottom of the Options window.



Context

Contexts play an organizational role within an Asterisk dialplan and also define scope. You can view contexts as a way to keep different parts of the dialplan separate. This

comes handy for providing different reception destinations for different companies that share the same Asterisk server. Any call that Asterisk handles will begin in a certain context. The instructions defined in this context will determine what things may happen to the call. With this option you can change the context at which your IAX account is working.

Voicemail extension

Enter the extension at which you would like to check for new voicemail messages.

Register on startup

This option is for automatically registering the current account each time ZOIPER starts up.

Don't play ringbacktones

Tick this checkbox to mute all ringbacktones.

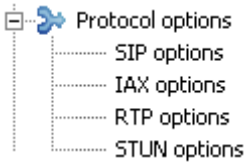
Custom codecs

You can choose to save the customised selection and order of codecs for this account. The Audio codecs for the chosen account are handled quite like in the general Audio codecs.

Do not forget to apply all changes by clicking on the Apply button. After applying the changes, the Audio codecs options for the current account appear underneath its name in the *Options Tree*.

Protocol options

You can enable/disable the Advanced Protocol options by ticking/unticking the Advanced options checkbox at the bottom of the ***Options*** window.



SIP options

Port

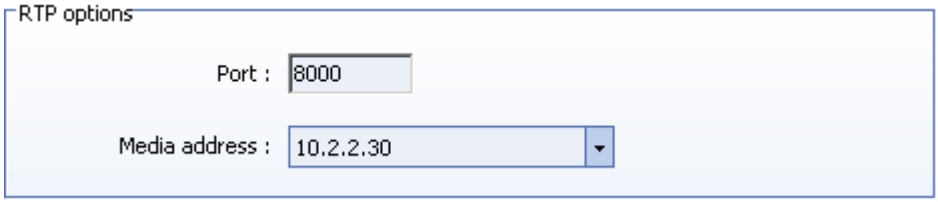
You can change the default port that SIP is using. The default port for SIP is 5060 as shown in the example above.

IAX options

Port

You can change the default port that IAX is using. The default port for IAX is 4569 as shown in the example above.

RTP options



Port

You can change the default port that RTP is using. The port number could range from 8000 (default) to 8100.

Media address

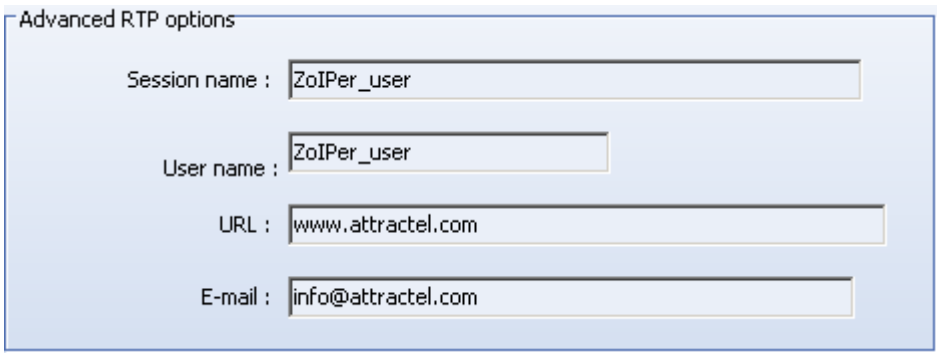
The Media address is negotiated by the SIP in order for the RTP to follow the correct address. The default Media address is the external IP of your network.

On startup ZOIPER tries to select the external IP. In case when no external IP is present, ZOIPER selects the internal IP.

Do not forget to apply all changes by clicking on the Apply button.

- *Advanced RTP options*

You can enable/disable the ***Advanced RTP options*** by ticking/unticking the Advanced options checkbox at the bottom of the ***Options*** window.



Session name

Enter Session name a name for all the RTP sessions.

User name

Enter your preferable User name.

URL

Enter your URL.

E-mail

Enter your E-mail address.

STUN Options

Enable STUN

Tick this checkbox if you have a STUN server.

Server Hostname/IP

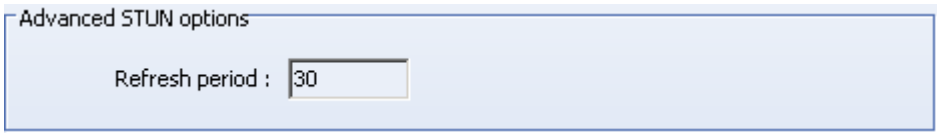
Enter the IP address of your STUN server or the IP address given to you by your VoIP Service Provider.

Port

You can change the default port that STUN is using. The default port for STUN is 3478 as shown in the example above.

- *Advanced STUN options*

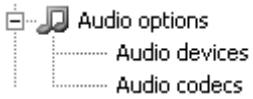
You can enable/disable the ***Advanced RTP options*** by ticking/unticking the Advanced options checkbox at the bottom of the ***Options*** window.



Refresh period

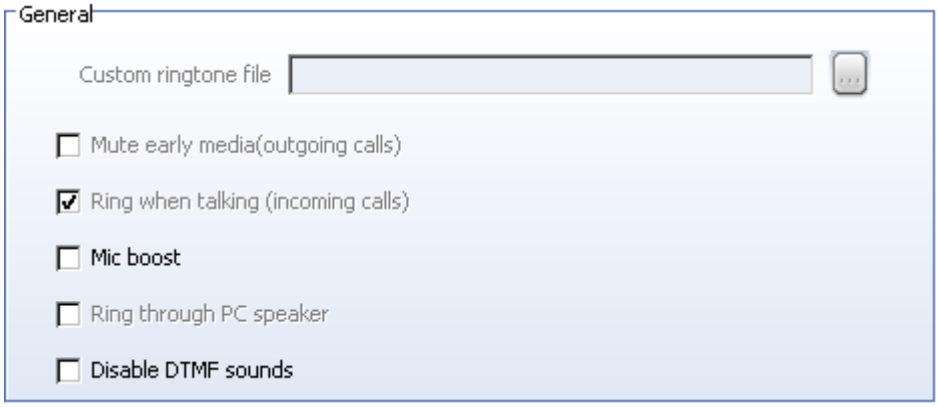
You can change the refresh period (in seconds) for the STUN server. The initially set value is 30 seconds as in the example above.

Audio options



General

You can adjust the General audio options. Always confirm your settings by clicking Apply.



Custom ringtone file

You can browse to a preferred ring tone or enter the file path. A ring tone file must be an 8 kHz 16-bit Mono wave file (.wav).

Mute early media(outgoing calls)

You can choose to enable/disable hearing ringtone when you dial a call. Also blocks early media.

Ring when talking (incoming calls)

You can choose to enable/disable the ringtone, while in a middle of a conversation.

Mic boost

To turn up the volume of your microphone, tick this option. Please mind that this might negatively affect sound quality.

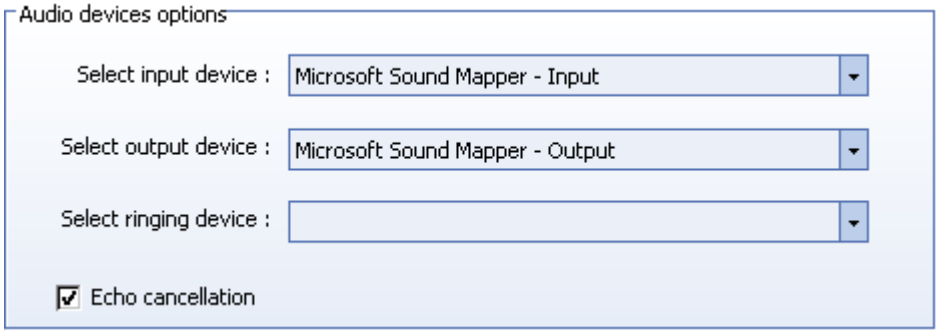
Ring through PC speaker

Tick this option if you want to use a beep from your PC speaker. Please mind that this does not mute the ringing through your headphones or speakers.

Disable DTMF sounds

Tick this option if you do not want to listen to sounds of DTMF buttons pressed.

Audio devices



Select input/output/ringing device

You can select your audio input/output/ringing device(headphones) from each corresponding dropdown menu. Please mind that the drivers for these devices' must be properly installed and recognized by Windows.

Echo cancellation

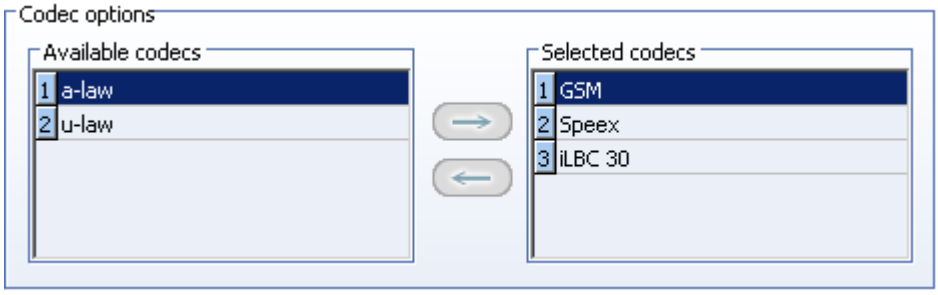
Tick this option in case of an echo tail to the speech.

- *Audio devices advanced options*

The Audio enhancement option uses post-processing filters to enhance the audio quality. It works better with lower quality sound cards.

Audio codecs

The Codec options look as follows:



You can choose among the following codecs:

- GSM
- Raw µ-law (G.711)
- Raw A-Law data (G.711)
- Speex Audio
- iLBC

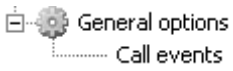
- These are the default settings for the audio codecs. They will be used by all accounts unless the custom codecs per account are used.

- If you want to use any of the *Available codecs* you have to select it and then press the right direction arrow.

- If you want to use stop using any of the *Selected codecs* you have to select it and then press the left direction arrow.

- Arrange the codec priority by dragging the blue numbers of the Selected codecs up and down. The codec with the lowest number has the highest priority.

General options



General

The window of the General options look as follows:

Start minimized

Tick this option in order for ZOIPER to start up minimised in your system tray.

Start with Windows

Tick this option in order for ZOIPER to automatically start up when opening Windows.

Always on top

With this option checked, ZOIPER windows will stay on your screen in front of all others.

Check for updates

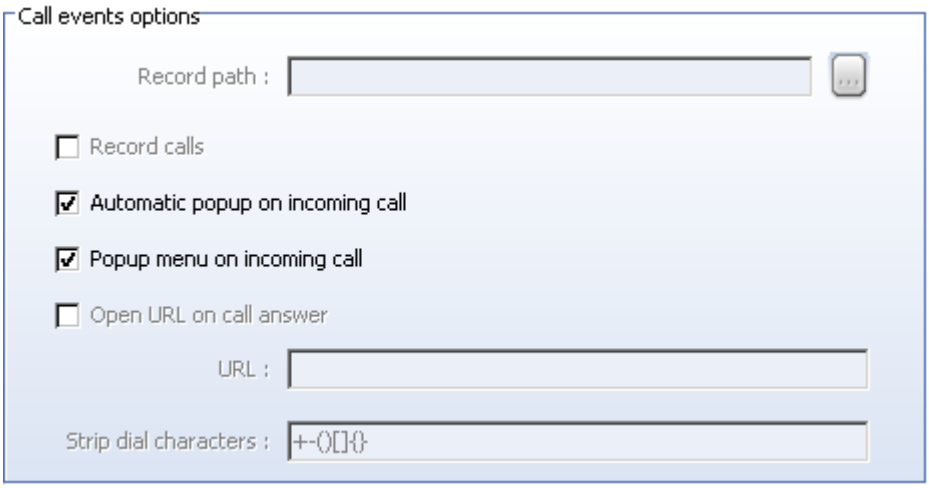
Tick this option in order for ZOIPER to automatically check for updates on start up.

Start with Windows

Tick this option in order for ZOIPER to automatically start up when opening *Windows*.

Call events

The Call events options look as follows:



Record calls, **Open URL on call answer** and **Strip dial characters*** are features from ZOIPER Biz. These features are not available with the ZOIPER Free version.

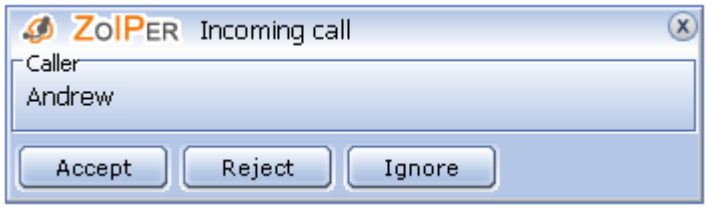
*Note that the +-()[]{} characters are stripped from extensions by default.

Automatic popup on incoming call

Choose this option if you want ZOIPER to pop up when there is an incoming call.

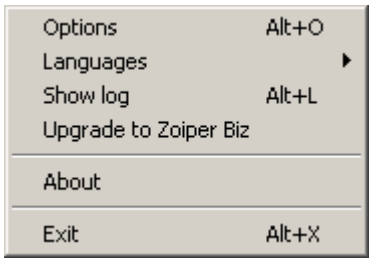
Popup menu on incoming call

When this option is checked, the following pop-up window appears every time you have an incoming call:



Languages

In order to use ZOIPER with your language, left-click on the main window and make your choice from the available languages.

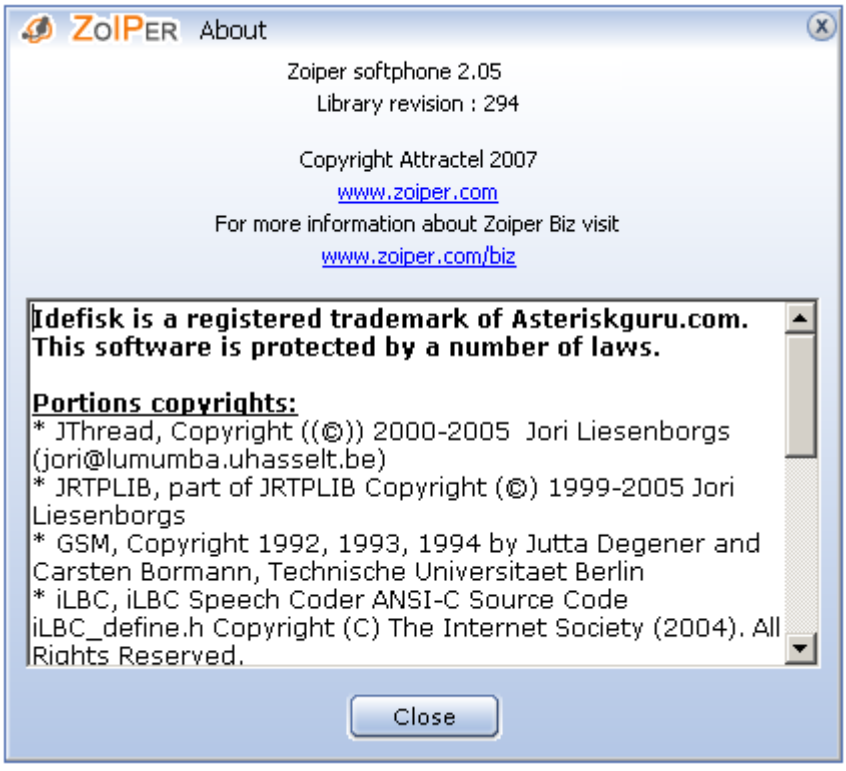


Upgrade to ZOIPER Biz

If you like the Free version of ZOIPER and wish to take advantage of the more advanced Biz version, just click on Upgrade to ZOIPER Biz in the window shown above.

About

On the screen ***About*** you can find more information about the version of ZOIPER installed on your computer.


ZoIPER About
Zoiper softphone 2.05
Library revision : 294
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* iLBC, iLBC Speech Coder ANSI-C Source Code
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Close